AMENDMENTS TO THE SPECIFICATION:

Page 1, before line 3, insert the following as a separate paragraph:

B2 --BACKGROUND OF THE INVENTION--.

Please amend the paragraph beginning at page 1, line 3 as follows:

B3 The present invention is concerned with the transmission of signals in discrete packets, and is especially concerned with the sending of audio signals, though it is also applicable to other kinds of signal, for example video signals._ More particularly it is concerned with the transmission of digitally coded audio signals in which information about successive frames of the audio signals is sent in successive discrete packets of a transmitted signal, which are then used by a receiver to create a replica of the original signal (for the purposes of discussion it will be assumed that there is a one-to-one correspondence between audio frames and transmission packets, though this is not actually essential). The invention seeks to address problems that arise when the transmitted information is lost or corrupted, so that one (or more) of the packets is unavailable to the receiver. Losses of this kind can occur in many types of transmission system, due for example to noise or (in a radio system) fading. In some types of system - for example connectionless services such as the Internet - different packets may be transmitted over different paths, and therefore be subject to different delays which can be so great as to result in the packets arriving in a different order from the order in which they were transmitted. Conventionally this is allowed for by providing the receiver with

a buffer which introduces a delay: the receiver stores the received packets in the buffer, and if the packets are numbered at the transmitter the receiver can then read the packets out of the buffer in the original sequence. For many applications this delay must be kept reasonably short if the overall transmission delay is not to be excessive, and the possibility remains that a packet may suffer a delay in excess of the buffer delay period. In such a case the packet is effectively lost, as the receiver is unable to make use of it. It has also been proposed (see, for example J. Bolot and A. Garcia, "Control Mechanisms for Packet Audio in the Internet", Proceedings of IEEE INFOCOM '96, Conference on Computer Communications, March 1996, pp 232-9 and V. Hardman, M. Sasse, M. Handley and A. Watson, "Reliable Audio for use over the Internet", Proceedings of INET '95, June 1995, pp 27-30. to provide redundancy in the signal, where each packet carries not only data pertaining to a frame of the audio signal but also data in respect of the previous frame of the audio signal, coded using a lower bit-rate coding algorithm, so that if a single frame is lost, this redundant data from the following frame can be decoded and used to fill in the gap that would otherwise occur in the decoded audio signal. However this process can be complex, and can give rise to difficulty due to discontinuous decoder operation, resulting in distortion.

Page 2, before line 5, insert the following as a separate paragraph:

--Brief Summary of Exemplary Embodiments--.

Page 4, before line 24, insert the following as a separate paragraph:

B5

--Brief Description of Drawings--.

Page 5, before line 5, insert the following as a separate paragraph:

B6

--Detailed Description of Exemplary Embodiments--.

Please amend the paragraph beginning at page 9, line 16 as follows:

B7

The samples are quantised according to bit allocations determined for each frame from an adaptive bit allocation procedure, to use the phenomenon of simultaneous masking to minimise the audible effects of sample quantisation._ Simultaneous masking occurs when a low-level signal component is made inaudible by a simultaneously occurring stronger component at some nearby frequency. A unit 43 applies a fast Fourier transform (FFT) to the signal, and supplies the result to a masking unit 44, where the masking properties of each audio frame are estimated (as described in the MPEG standard) using a psychoacoustic model and represented by a masking function *mask(k)* for the k'th sub-band (*k*= 0 ... 31). This masking function gives an estimate of signal level for sub-band *k* below which signals become inaudible or above which noise becomes audible. It is used to determine a signal-to-mask ratio *smr* for each of the 32 sub-bands:

$$smr(k) = sig(k) - mask(k)$$

where *sig(k)* is the signal energy within sub-band *k*. All these quantities are expressed in dB.

Please amend the paragraph beginning at page 18, line 1 as follows:

~~CLAIMS~~What is claimed is:.